REMARKS

Claims 1-47 are pending. Claims 1-47 have been rejected. Claims 1-47 remain in the case for reconsideration. Reconsideration is requested. No new subject matter has been added.

Claim Rejections – 35 U.S.C. § 102

Claims 13, 22 and 23 are rejected under 35 U.S.C. 102(b) as being anticipated by Gitlin et al. (US 5,856,971). The Examiner states that Gitlin dynamically adjusts the bit rate of a radio transmitter in response to a user input.

Claim 13 has been amended and now specifies an input detecting a user response requesting a different user perceived sound quality for a call and a controller configured to dynamically vary adaptation parameters used for transmitting packets making up the call to correspond with the requested user perceived sound quality in the user response.

This is clearly shown in all the drawings and particularly in FIG. 1 where a user 32 varies a user perceived sound quality heard over a handset 34. The user 32 listens to the perceived sound quality of a call 22 and then uses the call adaptation system 16 to vary the user perceived sound quality.

Neither Gitlin or any of the other cited art suggests varying adaptation parameters to correspond with a user perceived sound quality selection. Gitlin simply increases the bit rate that packets are transmitted from a mobile unit 200 to a base station 290. This has nothing to do with varying the adaptation scheme used for transmitting packets making up a call to correspond with user perceived quality requests.

Simply increasing the bit rate does not equate to improved user perceived sound quality. For example, transmitting data at a higher bit rate to the base station 290 may increase bandwidth bottle necks at the base station 290. As specified in the present invention on page 7, line 25: "Referring briefly back to FIG. 1, to improve voice quality, it may be tempting to simply "crank up the gain" by injecting a higher packet rate into the VoIP call 22. However, this can actually increase congestion in the packet network 26 and be counter-productive to improving audio quality."

That is the problem that could be caused in Gitlin by increasing the bit rate. Gitlin does not provide any way to identify a user perceived sound quality or vary call adaptation parameters to correspond with a user perceived sound quality selection. Gitlin simply increases a transmission bit rate that may or may not have any effect on the user perceived sound quality of a call.

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Claim Rejections – 35 U.S.C. § 103

Claims 1, 6, 9, 11, 24, 29, 32, 36, 41 and 44 are rejected under 35 U.S.C. 103(a) as being unpatentable over Gitlin in view of the Admitted Prior Art. Claims 2-5, 25-28 and 37-40 are rejected under 35 U.S.C. 103(a) as being unpatentable over Gitlin in view of APA and further in view of Hanley (US 6,198,910). Claims 12, 35 and 47 are rejected under 35 U.S.C. 103(a) as being unpatentable over Gitlin in view of APA and further in view of Albal (US 6,668,046). Claims 34 and 46 are rejected under 35 U.S.C. 103(a) as being unpatentable over Gitlin in view of APA and further in view of Le (US 6,300,887). Claim 14 is rejected under 35 U.S.C. 103(a) as being unpatentable over Gitlin in view of Skemer. Claims 15-17 are rejected under 35 U.S.C. 103(a) as being unpatentable over Gitlin in view Hanley. Claims 18 and 19 are rejected under 35 U.S.C. 103(a) as being unpatentable over Gitlin in view Shaanan. Claim 20 is rejected under 35 U.S.C. 103(a) as being unpatentable over Gitlin in view of Albal. Claim 21 is rejected under 35 U.S.C. 103(a) as being unpatentable over Gitlin.

First of all, Gitlin, Hanley, Shaanan, and some of the other art cited by the Examiner is related to cellular telephone networks. Cellular transmissions described in these patents are point to point transmissions that have a fixed bit rate that is maintained throughout the entire call.

Conversely, the IP calls specified in claims 1, 6, 9, 11, 24, 29, 32, 36, 41 and 44 use packet switched connections that have transmission rates and dropped packet rates that vary depending on network usage. IP networks have been primarily used for transmitting data and operate substantially different from the cellular network identified in Gitlin, Hanley, and Shaanan. Therefore, Gitlin, Hanley, and Shaanan should not be considered analogous prior art to the inventions described in claims 1, 6, 9, 11, 24, 29, 32, 36, 41 and 44.

Regardless, there are no suggestion in the Gitlin, Hanley, or Shaanan, or in any of the other cited prior art, that suggests dynamically adjusting the sound quality of a VoIP call to correspond to a user perceived sound quality selection. The Examiner refers to the Real Time Control Protocol (RTCP) as identifying voice quality indicators. However, no VoIP protocol has ever been used for dynamically adjusting a quality of a VoIP call in a packet switched network according to user perceived sound quality responses. This has the unexpected advantage of allowing the adaptation parameters to be adjusted to more effectively adapt sound quality to user perceived sound quality requests.

For example, if a user selects increased sound quality even after the system has already increased the packet rate for the call, the adaptation parameters may be modified to

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instead vary a codec that may encode more data in the same packet rate. This modification of the adaptation parameters could not be varied without the benefit of the user perceived sound quality feedback as specified in claims 1 and 13.

Simply using statistical packet rate information or packet drop information may not effectively adjust a sound quality level of a VoIP call over a IP packet switched network. Primarily, because there are such a wide variety of factors in IP network that effect user perceived sound quality.

Claim 2, 25, and 37 include the limitation of initially transmitting the packets in the VoIP call over an Internet Protocol (IP) packet switched network using an IP packet best effort transmission scheme and requesting reservation of IP packet switched network resources during the already established VoIP call when an increased sound quality request is detected from the user response

The Examiner first indicates that Gitlin should be considered a "best effort" method of transmission. This position is respectfully traversed. As mentioned above, Gitlin is not a best effort communication system. The wireless signaling sent between the mobile unit 200 and the base station 290 in Gitlin is a point to point transmission that, once established, provides a constant channel bit rate. Conversely, IP best effort transmissions have a transmission bit rate and packet drop rate that can vary according to bandwidth usage of other devices on the packet network.

Claims 2, 25, and 37 include the additional limitation of switching from the best effort transmission scheme to requesting reservation of network resources during the already established VoIP call when an increased sound quality request is detected from the user response. Claim 2 also includes the limitation of requesting reservation of IP packet switched network resources prior to the reserved IP packet switched network resources being used during the VoIP call and without necessarily using the entire requested resources during the VoIP call. Claims 4, 27, and 39 further specify switching from the best effort transmission scheme after the IP network resources have been reserved and the user response requests additional increases in the sound quality. This is clearly shown in FIG. 3 where the reservation resources are not necessarily used until the resources reservation is granted and the user selects a certain sound quality.

The Examiner states that Hanley monitors a request to increase sounds quality and requests reservation of network resources during the already established call. This is also respectfully traversed. Hanley does not request reservation of network resources as specified in claim 2. Conversely, Hanley as specified in claims 8 and 9 adjusts one of a plurality of

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parameters relating to the RF communication when the call quality value is less than said threshold minimum quality. This does not reserve the resources prior to actually using the network resources as specified in claim 2.

The problem with Hanley is that actually increasing the power level for each device below some threshold minimum quality value may cause some other devices to start operating below the threshold quality value. By reserving network resources prior to actually using the network resources, the call can be more certain of getting a specified sound quality even before the call actually uses those resources. However, those resources can be used for other calls when not being used by the call reserving the resources. Thus, there won't be a snowball effect of low quality calls when other calls request improved sound quality.

Regarding claims 3, 26, 28, the Examiner acknowledges that Gitlin does not disclose resource reservation using RSVP but that it would be obvious to implement this feature into Gitlin. The Examiner's position is respectfully traversed. The RSVP protocol is typically conducted during the initial establishment of the call. Using a pre-call establishment protocol to reserve network resources according to a user perceived sound quality request is an unobvious use of the RSVP protocol and provide substantial unexpected results as described above.

Claims 10, 33 and 45 are rejected under 35 U.S.C. 103(a) as being unpatentable over Gitlin in view of APA and further in view of Skemer et al. (US 6,570,849). The Examiner acknowledges that Gitlin does not disclose monitoring congestion in a network used for conducting a call and varying adaptation schemes according to the monitored congestion. However, the Examiner states that Skemer discloses a system where network congestion is monitored in order to adapt VoIP calls to the changing network congestion.

Claim 10 has been written in independent form and specifies monitoring congestion in a network used for conducting the VoIP call and varying the adaptation schemes according to the user response to the user perceived sound quality and the monitored congestion.

This is not even remotely suggested in Gitlin or Skemer. Skemer at column 4 describes selecting an optimum segment size for transferring data across a network. In this method, the cost of transferring a segment equal in size to the MTU is sequentially compared to the cost of transferring segments with integer incremented multiples of the MTU.

The problem with this technique, as with all the other techniques described above, is that the transmission scheme is varied without any user perceived sound quality response. In packet switched networks, simply varying packet segment size does not necessarily vary the sound quality of the call. For example, when there is high bandwidth usage in the network.

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Conversely, the present invention as specified in claims 10, 33 and 45 takes into account both user perceived sound quality and the monitored network congestion when varying the adaptation schemes. For example, the network congestion monitoring may indicate a current high congestion condition. If the user response requests increased sound quality during the monitored high congestion condition, a non-bandwidth affecting adaptation parameter may be used to increase the user perceived sound quality. For example, another codec may be user that encodes more audio signal into a same packet size. This prevents generating more packets, further increasing network congestion, and possibly having an opposite adverse sound quality result.
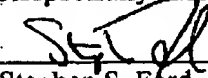
On the other hand, if the monitored network congestion is relatively low but the user response still requests increased sound quality, the adaptation parameters may be modified to increase the rate that packets are transmitted for the call.

Thus, both the user perceived sound quality responses and the monitored network congestion are used together to more effectively improve user perceived sound quality in a packet network.

CONCLUSION

For the foregoing reasons, reconsideration and allowance of claims 1-47 of the application as amended is solicited. The Examiner is encouraged to telephone the undersigned at (503) 222-3613 if it appears that an interview would be helpful in advancing the case.

Respectfully submitted,

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